"input start time". Claim 8 is dependent on claim 6. Claim 6 contains the step of determining, relative to the output audio signal, an input start time of the start of the input speech signal. Hence, when claim 8 refers to "the step of determining the onset marker", it should instead refer to "the step of determining the input start time" to be consistent with claim 6. This change is also consistent with the second line of claim 8 which also refers to "determining the input start time." This change is to correct an editing error and does not change the scope or meaning of the claim or add new matter to the specification.

Claim 34 has been amended to correct a typographical error. The claim was originally misnumbered as claim 340. The number of the claim has been changed from 340 to 34.

Rejection Under 35 U.S.C. 112

Claims 33 and 40 stand rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which Applicant regards as the invention. The examiner objected to the use of the phrase "further function" as being unclear and for having insufficient antecedent basis for the limitation "further function".

Claims 33 and 40 have been amended to clarify the claim language. The term "further function" is replaced with "also" and several resulting grammar changes have also been made. These changes remove any possible ambiguity in the claims and as a result the claims should be in condition for allowance.

The changes to claims 33 and 40 are made merely to correct awkward language. The changes do not change the scope of the claims and do not introduce any new matter.

Rejections Under 35 U.S.C. 102(b)

Claims 1-55 stand rejected under 35 U.S.C. 102(b) as being anticipated by Nguyen (U.S. Patent No. 5,765,130).

Nguyen is directed to a voice recognition system that includes a barge-in detector for determining the presence of an user input signal at the same time an output audio signal (prompt

signal) is being emitted by the system (col. 4 lines 46-49). The system provides an output audio signal to a telephone user (col. 4 lines 10-14). An echo canceller is used to remove most of the output audio signal that is fed back into system by the telephone (col. 4 lines 26-29). After echo cancellation a residue of the output audio signal remains in the signal fed from the telephone to the system (col. 4 lines 35-38). The barge-in detector has elements for sampling and calculating the energy of the output audio signal sent to the telephone and the signal received from the telephone (col. 4 lines 53-56 and col. 5 lines 60-63). These calculated energies are used to calculate an attenuation factor by which the output audio signal is attenuated after it is fed back by the telephone and passed through the echo canceller (col. 4 lines 53-59). The attenuation factor is used to estimate a replica signal which is an estimate of the portion of the signal received back from the telephone that is not part of the user input signal (col. 5 lines 57-63). The replica signal is compared to the signal received from the telephone to detect the presence of the user input signal (col.6 lines 47-50). When the user input signal is detected, the speech recognition unit, which is connected directly to the barge-in detector, is turned on and possibly the output audio signal is turned off. (col. 4 lines 63-67, FIG. 1). No mention is made in Nguyen of referencing the detected start of the user input signal to the output audio signal. Hence Nguyen suffers from the defect found in prior art barge-in detectors of not taking into account the uncertain delay characteristics found in wireless and packet data systems (Specification page 2 lines 2-5 and 21-28). These delays make it difficult to determine the portion of the output audio signal to which a user input signal corresponds.

Claim 1 is directed to a method for processing an input speech signal during presentation of an output audio signal including the steps of detecting the start of an input speech signal; determining, relative to the output audio signal, an input start time of the start of the input speech signal; and providing the input start time for use in responding to the input speech signal. Claim 1 is distinguishable from Nguyen for at least the reason that Nguyen does not determine an input start time relative to the output audio signal. The barge-in detector of Nguyen does detect the start of an input signal. (col. 4 lines 56-61). However, Nguyen does not reference the start of the detected input signal to the output audio signal. Instead Nguyen simply detects the input signal and notifies the speech recognition unit to turn on (col. 4 lines 61-62). Hence Nguyen does not determine an input start time of the input signal relative to the output audio signal. Because Nguyen does not calculate such an input start time referenced to an output audio signal, Nguyen

can not provide the input start time for use in responding to the input speech signal. Hence claim 1 is allowable. Claims 2-5 are dependent on claim 1 and are therefore allowable for at least the same reasons as claim1.

Claim 2 further limits claim 1 so that the input start time comprises any one of a time stamp relative to a temporal context of the output audio signal, a sample index relative to a sample context of the output audio signal, and a frame index relative to a frame context of the output audio signal. Nguyen does not teach of using time stamps, sample indexes, or frame indexes for any purpose. Hence claim 2 should be allowed. Claim 3 is dependant on claim 1 so should be allowed.

Claim 4 is directed to a method for processing an input speech signal during presentation of an audio output signal including the steps of detecting the start of an input speech signal; determining an identification corresponding to the output audio signal; and providing the identification for use in responding to the input speech signal. Claim 4 is distinguishable from Nguyen for at least the reason that Nguyen does not determine an identification corresponding to the output audio signal. Hence Nguyen can not determine an identification or provide the identification for use in responding to the input speech signal. Claim 4 is therefore allowable. Claim 5 is dependant on claim 4 and so should also be allowed.

Claim 6 is directed to a method for processing an input speech signal in a subscriber unit of a wireless communication system including the steps of detecting the start of an input speech signal during presentation of an output speech signal; determining, relative to the output audio signal, an input start time of the start of the input speech signal; and providing the input start time to a speech recognition server as a control parameter. Claim 6 is distinguishable from Nguyen for at least the reasons that Nguyen does not determine an input start time relative to the output audio signal . The barge-in detector of Nguyen does detect the start of an input signal. (col. 4 lines 56-61). However, Nguyen does not reference the start of the detected input signal to the output audio signal (prompt signal). Instead Nguyen simply detects the input signal and notifies the speech recognition unit to turn on (col. Lines 61-62). Hence Nguyen does not determine an input start time of the input signal relative to the output audio signal. Because Nguyen does not calculate such an input start time referenced to the output audio signal, Nguyen

can not provide the input start time to the speech recognition server as a control parameter. Furthermore, Nguyen does not detect the presence of an input speech signal within a subscriber unit as in claim 6 (FIG. 1). Hence claim 6 is allowable. Claims 7-12 are dependent on claim 6 and should therefore also be allowable.

Claim 7 adds to the method of claim 6 the step of receiving at least one information signal from the speech recognition server based at least in part upon the input start time. As discussed above with regard to claim 6, Nguyen does not calculate such an input start time (i.e. one referenced to the output audio signal) and hence can not receive an information signal based on at least the input start time. Therefore claim 7 should be allowed.

Claim 8 adds to the method of claim 8 the step of determining the input start time no earlier than the start of the output audio signal and no later than a start of a subsequent output audio signal. As discussed with regard to claim 6, Nguyen does not teach of determining such a start time (i.e. relative to the output audio signal), but even if it did teach determination of such a start time Nguyen would still not teach of determining the start time earlier than the start of the output audio signal and no later than a start of a subsequent output audio signal. Hence claim 8 should be allowed.

Claim 9 adds a limitation to claim 6 that the input start time is a time stamp, a sample index, or a frame index relative the output audio signal. Nguyen does not teach of using time stamps, frame indexes, or sample indexes so claim 9 should be allowable.

Claim 11 adds to claim 6 that the output audio signal comprises a speech signal synthesized by the subscriber unit in response to a control signal from the infrastructure. Nguyen does not teach a subscriber unit synthesizing speech. Claim 11 should therefore be allowed.

Claim 12 adds to the method of claim 6 the steps of analyzing the input signal to provide a parameterized speech signal; providing the parameterized speech signal to the speech recognition server; and receiving at least one information signal from the speech recognition server based at least in part upon the input start time and the parameterized speech signal. Nguyen does not teach of a subscriber unit analyzing a speech signal to provide a parameterized

speech signal. Therefore it can not provide parameterized speech or receive an information signal based in part on a parameterized speech signal. Hence claim12 should be allowed.

The Examiner rejected claim 13-30 as being method claims similar and scope and content to method claims 6-12. The examiner also rejected claims 31-55 as apparatus claims implementing methods similar in scope to claims 6-12. We respectfully disagree. While claims 13-55 contain much subject matter in common with claims 6-12, many of the claims contain additional novel features. For example, in claim 6 an input speech signal is detected within a subscriber unit during the presentation of an output speech signal. In claim 17, an information signal based at least in part on an identification and a parameterized speech signal is received by a subscriber unit. In claim 18, an input start time is received from a subscriber unit by a speech recognition server. In claim 21 information signals are directed to a subscriber unit to control the operation of the subscriber unit. In claims 22 and 28, a subscriber unit is coupled to at least one device and information signals are directed to the at least one device to control operation of the at least one device. These examples are merely to illustrate some of the additional novel features of claims 31-50 and are not meant to be an exhaustive list.

Attached hereto is a marked-up version of the changes made to the specification and claims by the current amendment. The attached page is captioned "Version with markings to show changes made".

Applicants respectfully request that a timely Notice of Allowance be issued in this case. The Examiner is invited to contact the below-listed agent if the Examiner believes that a telephone conference will advance the prosecution of this application.

Respectfully submitted,

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VERSION WITH MARKINGS TO SHOW CHANGES MADE

IN THE CLAIMS

8. The method of claim 6, the step of determining the <u>input start time</u> [onset marker] further comprising the steps of:

determining the input start time no earlier than a start of the output audio signal and no later than a start of a subsequent output audio signal.

33. The subscriber unit of claim 32, further comprising:

means for analyzing the input speech signal to provide a parameterized speech signal, wherein the means for providing [further function to] <u>also</u> provide<u>s</u> the parameterized speech signal to the speech recognition server, and the means for receiving [further function to] <u>also</u> receive<u>s</u> the at least one control signal from the speech recognition server based at least in part upon the input start time and the parameterized speech signal.

[340] <u>34</u>. The subscriber unit of claim 31, wherein the means for determining the input start time function to determine the input start time no earlier than a start of the output audio signal and no later than a start of a subsequent output audio signal.

40. The subscriber unit of claim 39, further comprising:

means for analyzing the input speech signal to provide a parameterized speech signal, wherein the means for providing [further function to] <u>also</u> provide<u>s</u> the parameterized speech signal to the speech recognition server, and the means for receiving [further function to] <u>also</u> receive<u>s</u> the at least one control signal from the speech recognition server based at least in part upon the identification and the parameterized speech signal.